mation in the communication devices **18**a, **18**b, **19**a, **19**b, to enable any corresponding audible signals that are generated to have amplitudes that are within a predetermined range of amplitudes (and thus be normalized accordingly). This procedure compensates for differences between the amplitudes of recorded sound waves and desired amplitude levels (and thus standardizes the volumes of generated sounds). Reference is now made to FIG. **5**, which illustrates a block diagram of a method in accordance with this embodiment of the invention. Step **300** of that diagram preferably is performed when an audible signal or acoustic information is entered into a user communication device, such as in step **122** of FIG. **3**a, step **224** of FIG. **4**a, and step **252** of FIG. **4**b. For the purposes of this description only, the method of FIG. **5** will be described in the context of being performed in response to party P**2** entering the audible signal **21**b' into the communication terminal **18**b in step **224**, although it should be noted that a similar procedure also may be performed in response to acoustic information being entered in one or more of the steps **122** and **252** as well, and the method may be performed in other devices **18**a, **19**a, **19**b besides terminal **19**b.

In step **300**, in response to the audible signal **21**b' being entered into the microphone **21**b of terminal **18**b, and eventually being converted to digital form by the A/D converter **21**a and provided to the controller **18** as acoustic information (as in step **224**), the controller **18** of the terminal **21**b performs a first predefined algorithm to compute one or more acoustic characteristics of the inputted signal, based on the digital values of the acoustic information received from A/D converter **21**a. Those characteristics preferably include amplitude information representing a maximum amplitude of the entered audible signal **21**b**1**', and a minimum amplitude of the entered signal **21**b', and are determined based on individual bits or words (e.g., 16 bits) included in the acoustic information, using any suitable, known algorithm. For example, the algorithm may be performed by examining each 16 bit word received in succession, maintaining a running tally of the minimum and maximum values of all the received words, and maintaining a running total of all values of the received words (i.e., each time a value of a next word is determined to be less than a current minimal value, or greater than a current maximum value, that new value is recorded as the new minimum or maximum value, and is added to the running total). The minimum and maximum values remaining after all of the words have been received are considered to represent the minimum and maximum amplitudes, respectively, of the entered signal **21**b'. The first predefined algorithm preferably is performed "on the fly", as a predetermined number of bits or words are received in the controller **2**a (prior to those bits being stored in the memory **24** as described above).

Thereafter, in step **302** the controller **18** performs a second predefined algorithm using both the maximum and minimum amplitude values determined in step **300** and predetermined information (pre-stored in memory **24**) representing desired maximum and minimum amplitude values, respectively, to determine a scaling factor to be used in scaling the digital values representing the entered signal **21**b'. For example, the second predefined algorithm may include the following:

$$\text{if } o\text{min}=o\text{max, then scaling factor}=1.0,$$

$$\text{else scaling factor}=(d\text{max}-d\text{min})/(o\text{max}-o\text{min});$$

wherein omin and omax represent the determined minimum amplitude and maximum amplitude, respectively, of the audible signal **21**b', and dmax and dmin represent the predetermined maximum value and predetermined minimum value, respectively.

Control then passes to step **304** where the controller **18** employs the determined scaling factor in a third predefined algorithm to scale the digital values (now stored in memory **24**) representing the entered signal **21**b', to cause the values to be placed within a range bounded by the predetermined maximum and minimum values. For example, according to one embodiment of the invention, the third predefined algorithm includes performing, for each individual word representing the audible signal **21**b', the following algorithm (ALS) for scaling the word:

$$w(i)=\text{scaling factor}*(w(i)-o\text{min})+d\text{min} \quad \text{(ALG)}$$

wherein w(i) represents an individual word, and omin and dmin are defined as described above.

As an example, assuming that the controller **18** determines in step **300** that the maximum amplitude value of the entered audible signal **21**b' is '150' and the minimum amplitude value of that signal is '0', and the corresponding predetermined maximum and minimum values are '300' and '0', respectively, then the performance of step **302** results in a determination that the digital values representing the entered signal should be scaled by a factor of '2', and step **304** is performed by multiplying those values by that factor '2'. In this manner, the acoustic information representing the entered signal **21**b' is normalized in the memory **24**. Thereafter, the method continues (in step **224**) in the above-described manner. As a result of the normalization procedure, when the acoustic information is later D/A-converted and outputted again as an audible signal the signal will have an amplitude which is within a predetermined range of amplitudes values (and a resulting sound volume will be within a predetermined range of volumes).

In accordance with another embodiment, the acoustic characteristics obtained in step **300** are stored in the memory **24** along with the inputted acoustic information, for subsequent use in normalizing the acoustic information when it is later retrieved for use in generating an audible signal. For example, in this embodiment the steps **300** and **302** are performed in a similar manner as described above. However, after the acoustic characteristics are determined in step **300**, they are stored in the same memory location as the entered acoustic information, and are subsequently included in the call signal later formed and transmitted (in step **230**). Thereafter, assuming control passes to step **254** of FIG. **4**b in the above-described manner, then the controller is of terminal **18**a (after storing the received acoustic information in the memory **24** in step **252**) performs step **254** by extracting the acoustic characteristics included in the received call signal (sent in earlier step **230**), performing the second predefined algorithm in the above-described manner, using those characteristics (the maximum, and minimum amplitude values) and predetermined information (pre-stored in memory **24** of terminal **18**a) representing desired maximum and minimum amplitude values, and by performing the third predefined algorithm in the above-described manner, based on the scaling factor determined as a result of the second predefined algorithm. The performance of the third predefined algorithm causes the scaling of the digital values (e.g., words) represented by the acoustic information now stored in memory **24** of terminal **18**a, and causes those values to be placed within the range bounded by the predetermined maximum and minimum values. Thereafter, step **254** is performed as described above, where the controller **18** provides the normalized information to the D/A converter **17**a, which then responds by outputting a corresponding analog signal to the speaker **17**. The speaker **17** then generates a corresponding audible signal indicating the receipt of the incoming call (step **254**). In this manner, the